EQUALIZER FOR PA SYSTEM

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ABSTRACT

In audio processing, equalization is the process of changing the frequency envelope of a sound. In passing through any channel, frequency spreading occurs. In this project I have build 9 band graphic equalizer. The graphic equalizer has become very popular in recent years. It is called graphic because, as the front-panel sliders or knob are adjusted; their positions give an approximate display of the resultant frequency response. This device divide the audible spectrum into nine frequency bands, and allow adjustments to each band via its own boost/cut control gain from 1.17 to 6. A growing number of audio enthusiasts are using equalizers to adjust their stereo system's frequency response, whether to compensate for room acoustics, for PA system or for creative recording purposes. Instead of broad adjustments of treble, bass, and maybe the midrange (sometimes called presence), we now have independent control over the low bass, mid-bass, high bass, low midrange, and so forth.

CHAPTER I

INTRODUCTION

1.1 Problem Statement

In order to throw out a simple gathering or function, the current practices and procedures applied is quite intensify as one needs to fill a form and wait for turn. Yet for a small gathering or a prompt function which need to be done in such a hurry, this will cause slightly a problem.

In order to unravel this problem and make things less complicated for all personnel involved, students from Faculty of Electrical Engineering will design PA system that consist of mixer, equalizer, and power amplifier that can be used for a small function or gathering.

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1.2 Introduction To Project

Engineering Project I, BEE 4172 is a partial requirement for Bachelor of Electrical Engineering. For this particular course, I've been given a task to construct, building and testing of an equalizer unit to be used as part of a public address (PA) system. The equalizer should have a minimum of five different channels for adjusting the frequency response of the input audio signal

The purpose of this project is to make the quality sound of a public address system better than before it use the equalizer. This can happen by cutting or boosting the gain of the output in the range of frequency. Equalizer (EQ) is a combination of a filter; usually adjustable, chiefly meant to compensate for the unequal frequency response of some other signal processing circuit or system. An EQ filter typically allows the user to adjust one or more parameters that determine the overall shape of the filter's transfer function. It is generally used to improve the fidelity of sound, to emphasize certain instruments, to remove undesired noises, or to create completely new and different sounds.

1.3 Objective

The objectives of this project are to;

i. To make a useable equalizer for faculty.
ii. To synchronize this device with mixer and power amplifier to make a low cost PA system for faculty.

1.4 Scope of Works

In this project I will focus on;

i. The designation of physical look of the equalizer.
ii. Fabrication of equalizer circuit.
iii. Testing the overall performance of this device.

CHAPTER 2

LITERATURE REVIEW

2.1 Introduction to Sound

Sound is wave that carries information from one point to another point as well as energy. In this thesis the focus is more on the longitudinal sound waves in air. Longitudinal means that the back and forth motion of air is in the direction of travel of the sound wave. The changing in air pressure allows us to hear which will be covered later in this chapter later on.

2.1.1 Source of Sound

There are so many sources of sound that we can find but in this thesis I will focus more on vibrating bodies and changing airflow. Vibrating bodies is a when something like guitar string or piano vibrates it displaces the air next to it and causes the air pressure to increase and decrease slightly. These air pressure fluctuations can be called as sound wave. This source of sound is the familiar source that we can find.

Changing airflow happens when we speak or sing and when our vocal folds alternately open and close so that the rate of air from our lung increases and decreases, resulting in a sound wave. We can find this situation when people use some music instruments such as like clarinet, saxophone or trumpet.

2.1.2 Noise

There are many sounds that we hear everyday but not all of that we wanted to hear. The unwanted sound we called as noise. In this thesis the discussion are more on one way to eliminate one noise that we called as acoustic feedback.

Acoustic feedback occurs when the amplified sound from any loudspeaker reenters the sound system through any open microphone and is amplified again and again and again.

There are so many way to eliminate acoustic feedback. It can be divided into two. Firstly one is by physical way including ask the talker to talk more louder, reduce the distance from the talker to the microphone, reduce the number of open microphones, and move the loud speaker farther away from the microphone.

Secondly is by using electrical way such as using an equalizer to cut the frequency band in which the feedback occurs or using frequency shifter to shift the frequency band that the feedback occurs.

2.2 Waves

As we already know that sound waves in air are longitudinal waves. Many other types of waves such as light waves and radio waves are transverse waves. Even these types of waves are different from sound waves but they have similar behavior.

The first thing about the waves is they can transport energy and also information from one place to another place through a medium.

The other properties are they can be reflected, refracted, or diffracted [1]. But they all travel with different speed in different medium. Example sound waves travel through air as a medium only 344m/s ≈ 600km/h but light can travel as fast as 3×10^8 m in 1 second.

2.2.1 Sound Waves

Sound waves can travel through many medium states such as solid, liquid, or gas. We can try to test the movement characteristic of sound waves by placing a large pipe or tube with the loudspeaker at one end. An electrical impulse to the loudspeaker causes the cone to move forward suddenly, compressing the air in front of it very slightly. This pulse of air pressure travels down the tube at a speed of about 340m/s. It may be absorbed at the far end of the tube, or it may ne reflect back toward the loudspeaker depending in what is at the end of the tube [1].

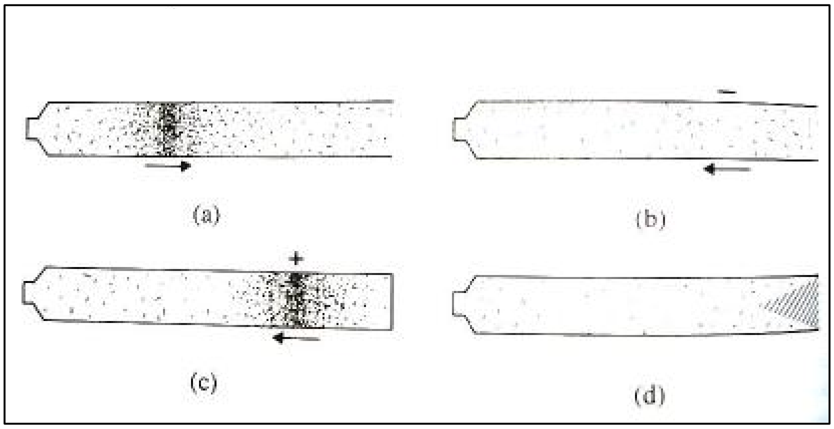

Figure 2.1: Reflection of the sound pulse in a pipe; (a) incident pulse; (b) reflection at an open end; (c) reflection at a closed end; (d) no reflection on the absorbing end.

Reflection of a sound pulse for a 3 different end condition is illustrated in Figure 2.1. If the end is open the excess pressure drops to zero, and the pulse reflects back as negative pulse of pressure as can be seen at Fig. 2.1 (b). If the end is closed the pressure build up to twice it value and the pulse reflects back as a positive pulse of pressure as can be seen in the Fig. 2.1 (c). If the end is terminated with a sound absorber, there is virtually no reflected pulse. Such a termination is called anechoic, which mean “no echo” [1].

The speed of sound can be known by using this formula

$$v = \sqrt{\frac{\gamma R T}{M}}$$

Where T is absolute temperature, M is the mass of the molecular gas and γ and R is the constant of the gas. The estimation value for M is $2.88 \times 10\text{-}2$, when R is 8.31 and γ is 1.4. By using all this value on that formula we can get $v = 20.1\sqrt{T}$. As it said before, T is absolute temperature by adding 273 and the Celsius scale. For an example if the temperature is 23 so T will become 300.

To make it simpler we can simplified the formula because the speed of sound increases by about 0.6m/s for each Celsius degree. So the new formula will be:

$$v = 331.3 + 0.6t \ m/s$$

Sound can travels faster in other medium such as liquid and solid. The speed of sound on several materials can be referred in Table 2.1.

Table 2.1: Speed of sound in several materials

Substance	Temperature (°C)	Speed (m/s)	Speed (ft/s)
Air	0	331.3	1,087
Air	20	343	1,127
Helium	0	970	3,180
Carbon dioxide	0	258	846
Water	0	1,410	4,626
Methyl alcohol	0	1,130	3,710
Aluminum	—	5,150	16,900
Steel	—	5,100	16,700
Brass	—	3,480	11,420
Lead	—	1,210	3,970
Glass	—	3,700–5,000	12–16,000

This table is taken from [1] where all the data here is subject to the result of an experiment made by the author. This thesis only takes this information for a review.

2.2.2 Doppler Effect

When the frequency of the source, f_s reach the observe, the frequency of the sound should be same as f_s but this only happen if neither source nor observer aren't moving. If they are moving toward each other, the observed frequency is greater than f_s; if they are moving apart, the observed frequency is lower than f_s. This apparent frequency shift is called the Doppler Effect [1].

The meaning of Doppler Effect can be made clearer by with the aid of a figure. From Figure 2.2, the center of the circle is source, S, and the observer, O at rest. If the source emits 100 waves per second, an observer at rest will receive the exact amount as the source emit. However a moving observer O' will receive more than 100 waves per second because it move toward source. A moving observer that moving towards will receive more because he or she "meets" the waves as he or she moves.

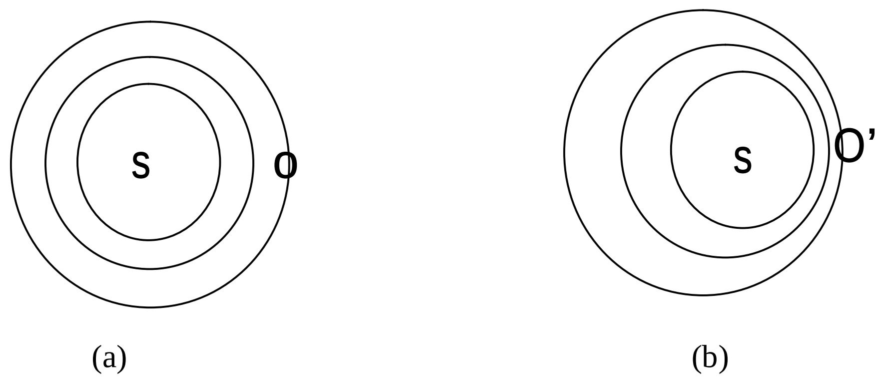

Figure 2.2 Doppler Effect (a) observer at rest (b) Moving observer

From the Figure 2.2 we can know the apparent frequency (the rate at which the observer meets waves). It will be

$$f' = f_s\left(\frac{v + v_o}{v}\right)$$

υ_o = the speed of the observer

υ = speed of sound

Note that if the observer passed the sound source, υ_o must be subtracted from the υ. Thus the frequency drop abruptly as the observer passes the source. There is another case if the source in motion. The observer will receive a greater rate than he or she would get from a stationary source. The apparent frequency will be

$$f = f_s\left(\frac{v}{v - v_s}\right)$$

But if the source moves directly toward observer, the frequency will drop abruptly, not gradually as the source pass by.

2.3 Frequency Response

As mentioned earlier, sound is cause by a vibration in air. This vibration will produce two things for sure. The first one is frequency and the second one is amplitude. Vibration occurs in a single wavelength and frequency is a measurement how much vibration occurs in a single second. It usually measured in Hertz (Hz). Frequency is directly correspondent to the pitch of sound. To make it clearer we can see from the Figure 2.3.

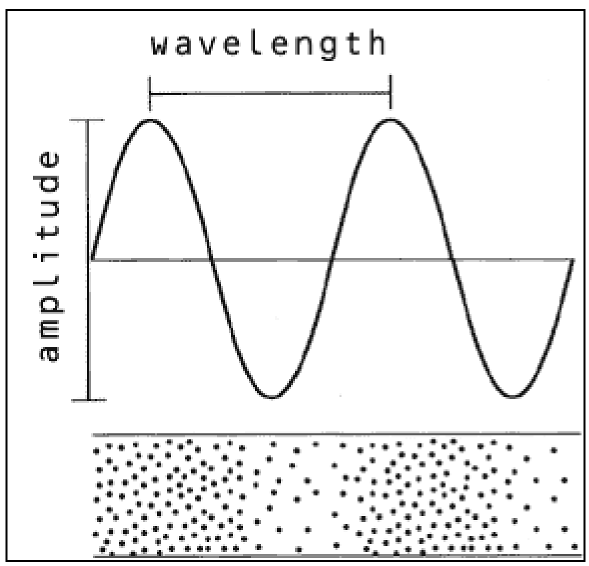

Figure 2.3: Two component in waves

Frequency response is the measure of any system's response at the output to a signal of varying frequency (but constant amplitude) at its input. It is usually referred to in connection with electronic amplifiers, loudspeakers and similar systems. The frequency response is typically characterized by the magnitude of the system's response, measured in dB, and the phase, measured in radians, versus frequency [2].

This project main focus is to have the desired frequency response by using an equalizer. The introduction to the equalizer will be discussed later on this chapter. The range of frequency that human can hear is from as low as 20 Hz till as high as 20 kHz. The desired frequency response is something like in Figure 2.4.

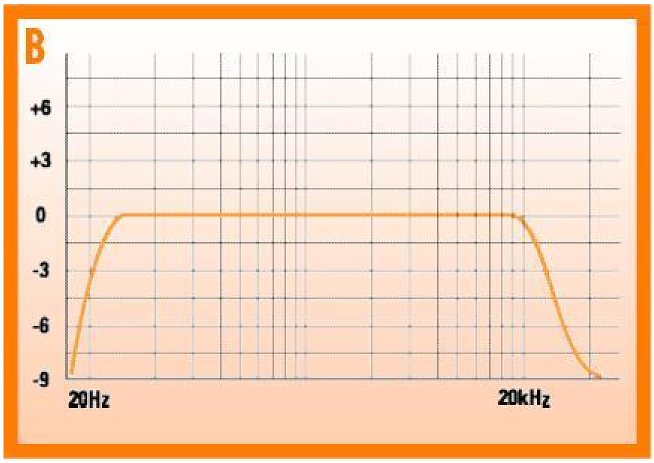

Figure 2.4: The desired Frequency Response

Figure 2.4 showed the ideal flat frequency response that mostly people want. But to achieve that in the real world is quite impossible even the flat response microphone has some deviation. More importantly, it should be noted that a flat frequency response is not always the most desirable option. In many cases a tailored frequency response is more useful. For example, a response pattern designed to emphasize the frequencies in a human voice would be well suited to picking up speech in an environment with lots of low-frequency background noise [3].

2.4 Human Auditory System

Human auditory system is a complex in structure and remarkable in function. Not only does it respond to a wide range of stimuli, but it precisely identifies the pitch and timbre (quality) of a sound and even the direction of the source. Much of the hearing function is performed by the organ named ear, but recent research has emphasized how much hearing depends on the data processing occurs in the central nervous system as well.

2.4.1 Range of Hearing

Most people will find that their hearing is most sensitive at 1-4 kHz and that is less sensitive at high and low frequency. Most of the children can hear frequency that a lot higher than what an adult can hear. That is why sometimes your children keep on complaining about sound that you cannot hear. People will loss their capability of hearing a higher frequency as they grow older. It is common for adults to have very low sensitivity for the highest frequencies.

It still debated issues about range of frequency that human can hear but it accepted worldwide that human can hear in range 20 Hz to 20 kHz. Sound below 20 is infrasonic and sound above 20 kHz is called ultrasonic. This range can be divided into 5 types called low bass, upper bass, mid-range, upper mid-range, and treble.

For the low bass (20 to 80 Hz) includes the first two octaves. These low frequencies are associated with power and are typified by explosions, thunder, and the lowest notes of the organ, bass, tuba, and other instruments. Too much low bass results in a muddy sound [4].

For the upper bass, (80 to 320 Hz) includes the third and fourth octaves. Rhythm and support instruments such as the drum kit, cello, trombone, and bass use this range to provide a fullness or stable anchor to music. Too much upper bass results in a boomy sound [4].

For the mid-range, (320 to 2,560 Hz) it includes the fifth through seventh octaves. Much of the richness of instrumental sounds occur in this range, but if over-emphasized a tinny, fatiguing sound can be the result [4].

For the upper mid-range, (2,560 to 5,120 Hz) is the eighth octave. Our ear is very particular about sound in this range, which contributes much to the intelligibility of speech, the clarity of music, and the definition or "presence" of a sound. Too much upper mid-range is abrasive [4].

For the treble, (5,120 to 20,000 Hz) includes the ninth and tenth octaves. Frequencies in this range contribute to the brilliance or "air" of a sound, but can also emphasize noise [4].

The range of sound intensity (pressure) and the range of frequency to which the ear responds are remarkable indeed. The intensity ratio between the sounds that bring pain to our ears and the weakest sounds we can hear is more than 1 trillion (10^{12}). The frequency ratio between the highest and the lowest frequencies we can hear is nearly 10^3 (1000) times, or more than nine octave (each octave represents a doubling frequency) [1].

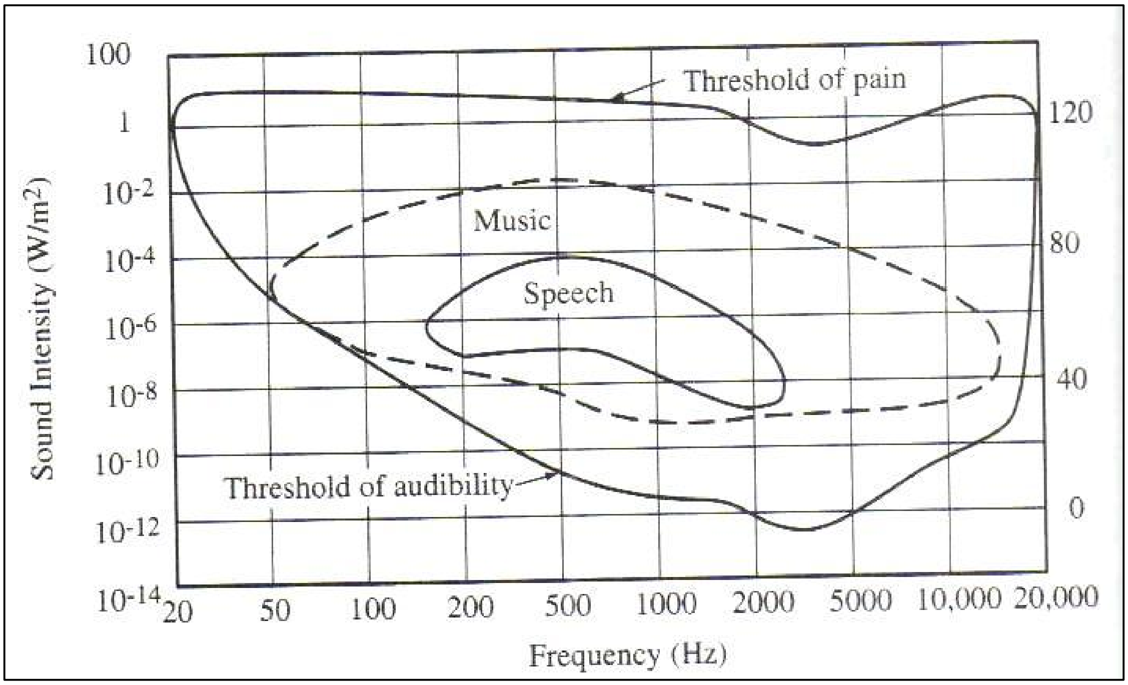

Figure 2.5: Range of frequencies and intensities to which auditory system responds [1]

From Figure 2.5 we can see how our auditory system responds to the intensity of sound. This figure is a result from experiment done by Fletcher and Munson (1933). The curve of threshold audibility demonstrates the relative intensity of the ear to sounds of low frequency at moderate to low intensity level.

2.4.2 Auditory System

As a human, our auditory system that is already known by everyone of us is our ears. Our ear can be divided into 3 parts which is first part is the outer ear, the second part is the middle ear, and the last part is the inner ear. Figure 2.6 below will show us what all that 3 parts are. This drawing is not to scale; for purposes of illustration, the middle ear and the inner ear has been enlarged.

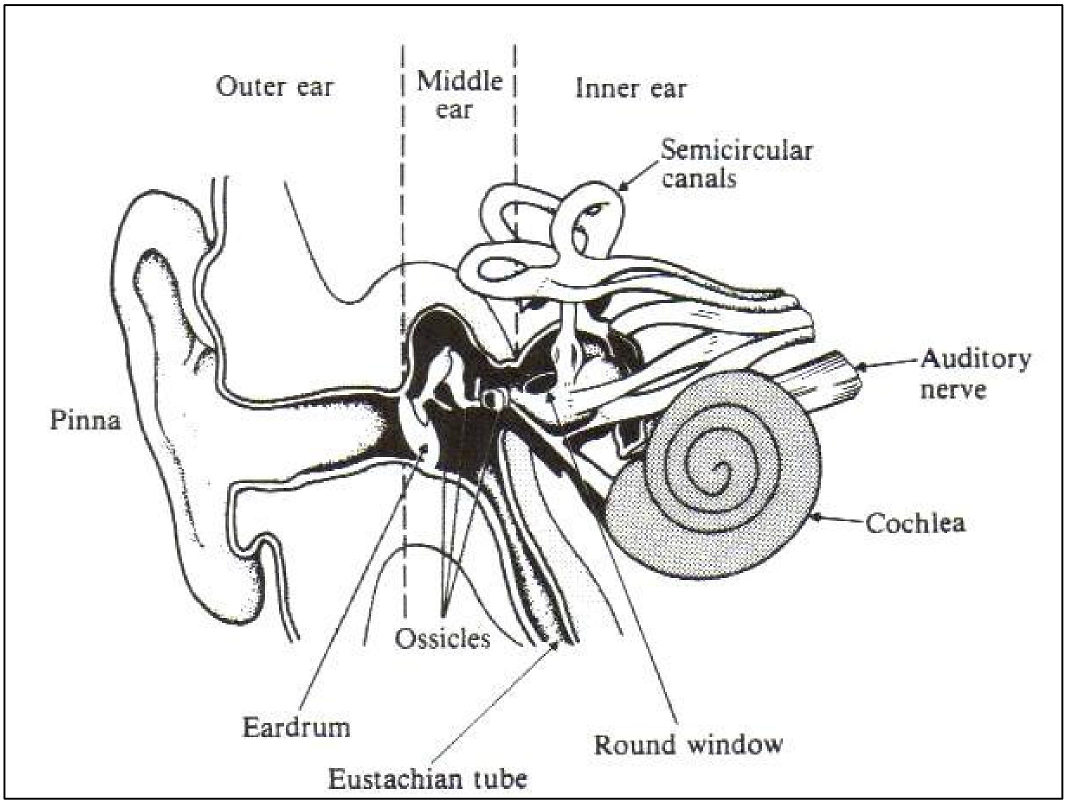

Figure 2.6: A schematic diagram of the ear, showing outer, middle, and inner regions

The outer ear consists of the external pinna and the auditory canal (meatus), which is terminated by the eardrum (tympanum). The pinna helps, to some extent, in collecting sound and contributes to our ability to determine the direction of origin of sounds of high frequency. The auditory canal acts as a pipe resonator that boosts hearing sensitivity in the range of 2000 to 5000 Hz.

The middle ear begins with eardrum, to which are attached three small bones (shaped like hammer, an anvil, and stirrup) called ossicles. The eardrum, which is composed of circular and radial fibers, is kept taut by the tensor tympani muscle. The eardrum changes the pressure a vibration of incoming sounds waves into mechanical vibration to be transmitted via the ossicles to the inner ear [1].

The function of ossicles is like a lever, which change the very small pressure to the much greater pressure up to 30 times. These 3 bones act like a mechanical transformer. The other function of ossicles is to protect the inner ear from very loud noises and sudden pressure changes. The responds to loud sounds is called acoustic reflex.

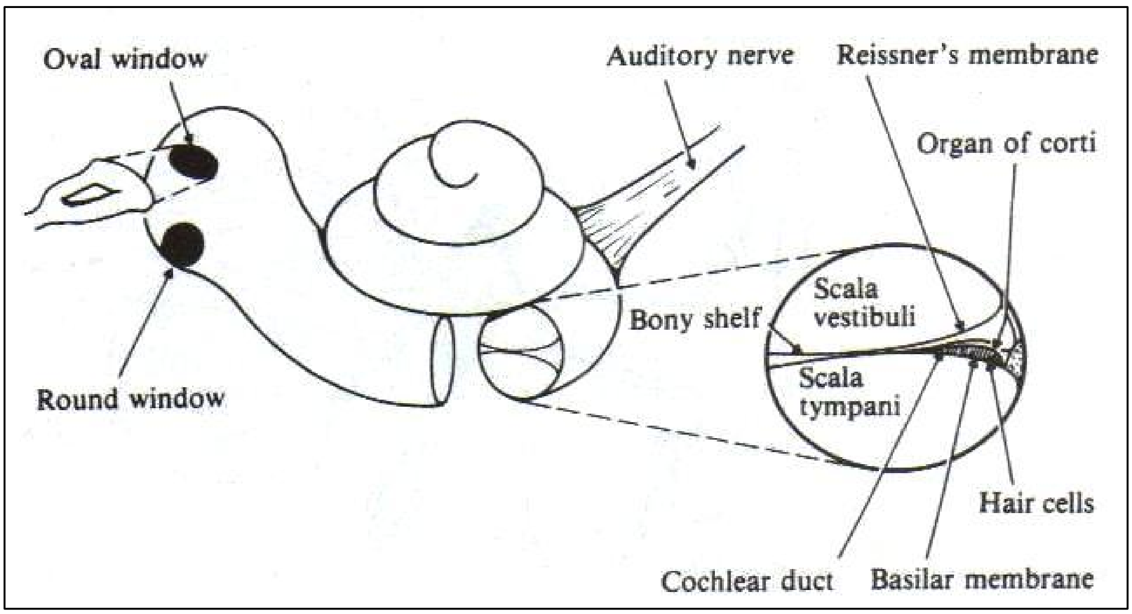

Figure 2.7: A schematic diagram of cochlea (inner ear) and a section cut of it

Inner ear as u can see from Figure 2.7 is the most complex part in the human auditory system. It contains the semicircular canals and the cochlea. Semicircular canals are for balancing of the body. It has nothing to do with the auditory system while cochlea is a crucial part of ear where it transforms pressure vibration to the properly coded neural impulse. Figure 2.8 below shows a schematic representation of the overall hearing mechanism.

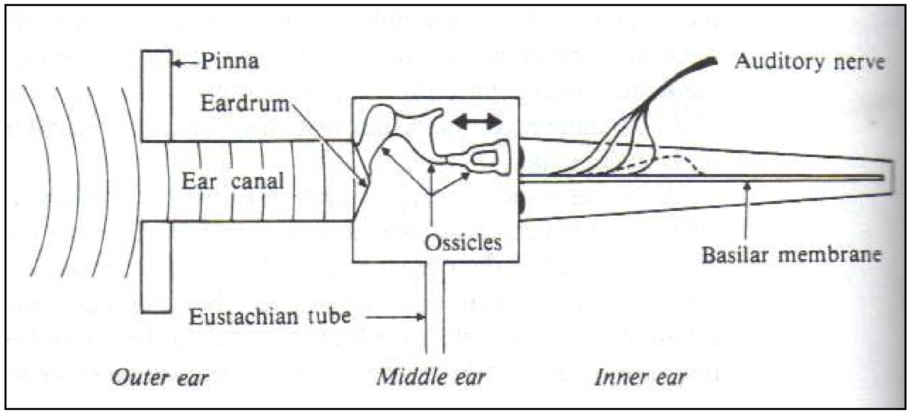

Figure 2.8: A schematic representation of the overall hearing mechanism [1].

2.4.3 Binaural Hearing and Localization

As a human we have been given 2 ears which we can have binaural hearing. The most benefit we can get from it is we can localize the source of sound. Binaural hearing really enhances our hearing ability by sense the direction of the sound. Lord Raleigh in 1876 performed an experiment and found that sounds of low frequency were more difficult to locate than those of high frequency [1].

2.5 Sound Pressure and Loudness

Most people might think it is imposible to measure the pressure and loudness of sound. However output signal of microphone usually is proportional to the sound pressure, thus sound can be measured with microphone and a voltmeter.

2.5.1 Decibels

Decibels scales are widely used to compare two quantities. We may express the power gain of an amplifier in decibels, or we may express the relative power of two sound sources. The decibels difference between two power levels, ΔL, is defined in term of their power ratio W_2/W_1:

$$\Delta L = L_2 - L_1 = 10\log\left({}^{W_2}/_{W_1}\right)$$

Although decibels scales always compare two quantities, one of these can be a fixed reference, in which case we can express another quantity in term of this reference. For example, we often express the sound power level of a source by using $W_0 = 10^{-12}$ W as a reference. Then the sound power level (in decibels) will be

$$L_W = 10 \log\left({}^{W}/_{W_0}\right)$$

2.5.2 Sound Intensity Level

Sound intensity level at a point some distance from the source can be expressed in decibels by comparing it to a reference intensity for which we generally use $I_0 = 10^{-12}$ W/m^2. Thus the sound intensity level at some location is defined as

$$L_I = 10\log\left({}^{I}/_{I_0}\right)$$

Even though sound power level and sound intensity level is measured in decibels but don't be confused with them. Sound power level is a power of a source of sound it self but sound intensity level is a measurement of sound at some point of distance from the source of sound. To further understand regardingt this, Figure 2.9 will help to show the different.

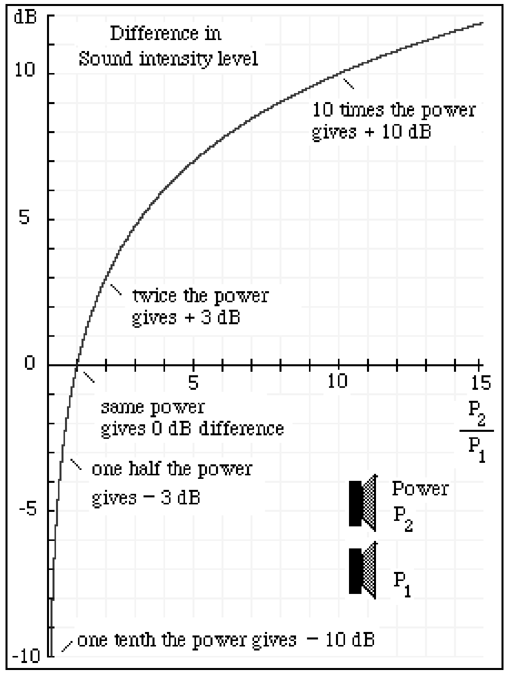

Figure 2.9: Graphically way to show sound intensity level measured in decibels

2.5.3 Sound Pressure Level

The intensity of a sound wave is proportional to the pressure squared. In other words, doubling the sound pressure quadruples the intensity. It can clearly be seen by this formula:

$$I = {p^2}/{\rho c}$$

I is the intensity of sound and p is a pressure where ρ is a density of air and c is a speed of sound. Sound pressure levels are measured by a sound-level meter, consisting of a microphone, an amplifier, and a meter that read in decibels. In the table below shows that the sound pressure levels of a number of sounds.

Table 2.2: Typical sound levels one might encounter

Jet takeoff (60 m)	120 dB	
Construction site	110 dB	Intolerable
Shout (1.5 m)	100 dB	
Heavy truck (15 m)	90 dB	Very noisy
Urban street	80 dB	
Automobile interior	70 dB	Noisy
Normal conversation (1 m)	60 dB	
Office, classroom	50 dB	Moderate
Living room	40 dB	
Bedroom at night	30 dB	Quite
Broadcast studio	20 dB	
Rustling leaves	10 dB	Barely audible
	0 dB	

Few questions might rise by looking to Table 2.2. Is it possible to have negative decibles? The answer to this question is; decibels are a ratio or difference between two quantities. So zero decibels means it is a reference level. In Table 2.2 the reference as one can see is a rustling leaves.

The sound level in dB is a measure (on a logarthmic scale) of the ratio of the sound pressure or sound intensity to this reference level. The logarithm of one is zero, so zero dB corresponds to the reference level. Numbers greater than one has positive logarithms, so positive a decibel means sound levels greater than that of the reference. Numbers smaller than 1 has negative logarithms, so negative decibel means sound levels below the reference level [5].

2.6 Speech Production

Of all creatures in this world, only human have the power of articulate speech. Speech is our chief means of communication. In addition, the human voice is our oldest musical instrument. The human voice and human ear are very well matched to each other. The ear has its' maximum sensitivity in the frequency range from 1000 to 4000 Hz, and that is the range of frequency in which the resonances of all vocal tracts occur.

2.6.1 The Vocal Organs

The aged thought that the only part of our body that related with our production of speech is our tongue and our mouth. The truth about production of speech, there is about more than ten parts of our body that contributing in production of speech. Figure 2.10 will show the overall vocal organs.

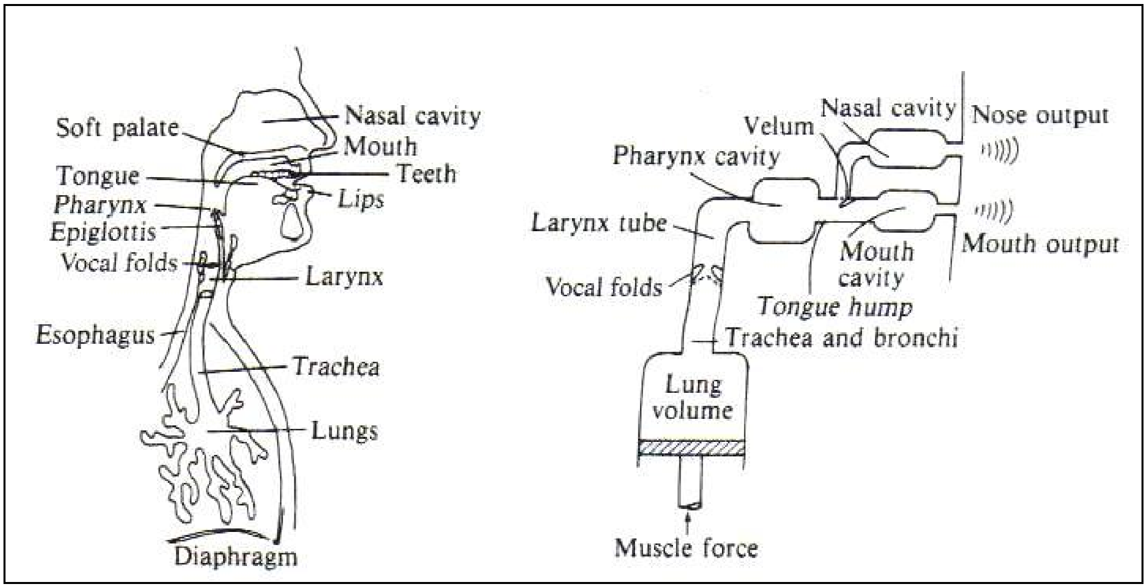

Figure 2.10: Human vocal organs and a representation of their main acoustical features.

The lungs serve as both reservoir of air and energy source. Whether in speaking or in exhaling, air is forced to lungs through larynx into three main cavities of the vocal tracts. These cavities are called pharynx, nasal and oral cavities. From the nasal and oral cavities, the air exits through the nose and mouth respectively.

In order to produce sound, the flow of air is interrupted by the vocal cords or by constriction in the vocal tract (made with tongue or lips). The sounds from the interrupted flow are appropriately modified by various cavities in the vocal tract and are eventually radiated as speech from the mouth and in some cases, the nose [1].

2.6.2 Articulation Speech

The articulation of English speech sounds also called phonemes is a basic thing to the speech. One or more phonemes combine to form a syllable and one or more syllable combine to form a word. It can be divided into two: vowels and consonants. Vowel sounds are produced with the vocal folds in vibration. Consonant may be either voiced or unvoiced. Table 2.3 will show some example of the vowels.

Table 2.3: The vowels of Great American English

Pure vowels					
ee	heat	/i/	aw	call	/ɔ/
i	hit	/ɪ/	ú	put	/ʊ/
e	head	/ϵ/	oo	cool	/u/
ae	had	/æ/	ǔ	ton	/ʌ/
uh	the	/ə/	er	bird	/ɜ/
ah	father	/ɑ/			

Various speech scientist lists from 12 to 21 different vowel sounds used in English language. This discrepancy in number comes about partly because of a difference of opinion as to what constitute a pure vowel sound rather than a diphthong. Figure2.11 below shows the approximate tongue position for articulating these vowels. Number 1-8 are the eight cardinal vowels, which serve as standard of comparison between languages [1].

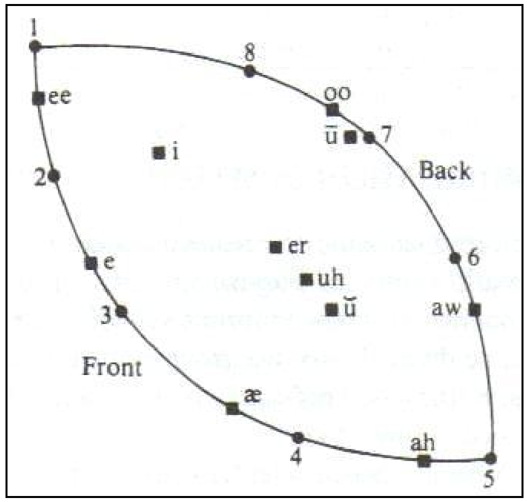

Figure 2.11: Approximate tongue positions for articulating vowels.

2.7 Equalizer

Equalizer (EQ) basiclly is a combination of many filters. These filters will be arranged according to the frequency band that it will able to control. There are many kind of equalizer that can be found in the market nowadays.

There are many kinds of EQ. Each one has a different pattern of attenuation or boost. A peaking equaliser raises or lowers a range of frequencies around a central point in a bell shape. A peaking equalizer with controls to adjust the level (Gain), bandwidth (Q) and center frequency is called a parametric equalizer. If there is no control for the bandwidth (it is fixed by the designer) then it is called a quasi-parametric or semi-parametric equalizer.

A pass filter attenuates either high or low frequencies while allowing other frequencies to pass unfiltered. A high-pass filter modifies a signal only by taking out low frequencies; a low-pass filter only modifies the audio signal by taking out high frequencies. A pass filter is described by its cut-off point and slope. The cut-off point is the frequency where high or low-frequencies will be removed. The slope, given in decibels per octave, describes how quickly the filter attenuates frequencies past the cut-off point. A band-pass filter is simply a combination of one high-pass filter and one low-pass filter which together allow only a band of frequencies to pass, attenuating both high and low frequencies past certain cut-off points.

Shelving-type equalizers increase or attenuate the level of a wide range of frequencies by a fixed amount. A low shelf will affect low frequencies up to a certain point and then above that point will have little effect. A high shelf affects the level of high frequencies, while below a certain point, the low frequencies are unaffected.

One common type of equalizer is the graphic equalizer, which consists of a bank of sliders for boosting and cutting different bands (or frequencies ranges) of sound. Normally, these bands are tight enough to give at least 3 dB or 6 dB maximum effects for neighboring bands, and cover the range from around 20 Hz to 20 kHz (which is approximately the range of human hearing). A simple equalizer might have bands at 20 Hz, 200 Hz, 2 kHz and 20 kHz, and might be referred to as a 4-band equalizer. A typical equalizer for live sound reinforcement might have as many as 24 or 31 bands. A typical 31-band equalizer is also called a 1/3-octave equalizer because the center frequencies of sliders are spaced one third of an octave apart [2].

2.7.1 Filter

To be able to construct equalizer, the most important part is the filter. The basic rule for equalizer is using filters. Filters can be divided to 4 types. The first one is high pass filter. The simple high pass filter is as shown in Figure 2.12.